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## (54) Adaptive, digital filter including a non-recursive part and a recursive part.

(57) An adaptive, digital filter including a non-recursive part and a recursive part, and which can be updated in a simple and reliable manner. The recursive part of the filter has a plurality of separate, permanently set recursive filters (13-16) with different impulse responses, and a linear combination is formed with adaptive weighting factors (W0-W3) from the output signals of the recursive filters (13-16). The filter is updated by a single signal (e(n)) being utilized for updating the non-recursive part (11) of the filter and the adaptive weighting factors (W0-W3) in the recursive part of the filter.

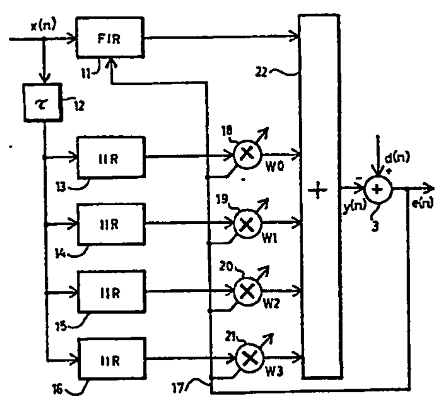


Fig.3

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## Description

# ADAPTIVE, DIGITAL FILTER INCLUDING A NON-RECURSIVE PART AND A RECURSIVE PART

### TECHNICAL FIELD

The invention relates to an adaptive, digital filter including a non-recursive part and a recursive part. Uses of the filter are such as echo canceller or equalizer in telecommunication equipment.

### BACKGROUND ART

The impulse response from a filter which is used for echo cancellation in telecommunication equipment shall as closely as possible immitate the impulse response of the transmission line in question. Included in the transmission line in such a case are two-wire to four-wire junctions, analogue-digital converters etc, which affect the impulse response. The latter generally has relatively long extension in time. It is therefore difficult to achieve a suitable impulse response with a filter which only has a finite impulse response. Such filters are called non-recursive filters or FIR filters (finite impulse response). For achieving a suitable impulse response, a filter for echo cancellation should comprise both a non-recursive part and a recursive part. Recursive filters are also called IIR filters (infinite impulse response).

There are known, reliable methods for updating adaptive FIR filters, i.e. adjusting the coefficients of such filters. They can be updated by minimizing the square of an error signal, which constitutes the difference between a so-called desired signal and the output signal of the filter. In such a case the desired signal may be a signal occurring on the receiver side in communication equipment where the filter is included. The square of the error signal can be minimized, e.g. according to the so-called LMS method (least mean square). The LMS method is described inter alia in the book: Widrow and Stearns, "Adaptive signal processing", pp 99-101.

Minimizing the square of an error signal according to the above is a so-called least square problem, due to the square of the error signal being a quadratic function of the filter coefficient values. This means that this square can be represented by a quadratic error surface, in an N-dimensional space where N is the number of coefficients, the optimum filter setting corresponding to the minimum point on this surface.

The corresponding square for an IIR filter is not represented by a quadratic error surface according to the above, however, and the error surface can have local minimum points instead. Known updating algorithms can fasten in such a local minimum point, resulting in that the optimum setting is never obtained.

Recursive filters can also be instable, as a result of that the poles in the Z transform of the transfer function can at least temporarily be moved outside the unit circle. For an IIR filter of the first degree, this means that the filter coefficient can be an amount greater than one, which makes the filter instable.

It is known to use a so-called "equation error" structure to avoid the problem with local minimii. In such a case two FIR filters are used, inter alia, of which one is connected to a transmitter side and the other to a receiver side in the same telecommunication equipment. An error signal is formed by the output signal of one filter being subtracted from the output signal of the other. The square of this error signal has a quadratic error surface, but a structure of this kind has the disadvantage that the minimized error signal does not represent the actual error. This is so, inter alia, when disturbances occur and when speech signals occur on the transmitter and receiver sides simultaneously. It has also been found difficult to adjust two filters which are connected in this way, due to the filters affecting each other. The equation error method is described, e.g. in the above-mentioned book "Adaptive signal processing", pp 250-253.

### DISCLOSURE OF INVENTION

The object of the present invention is to provide an adaptive digital filter which includes a non-recursive part and a recursive part, and which can be updated in a simple and reliable way. This is achieved by the recursive part of the filter having a plurality of separate, permanently set recursive filters with different impulse responses, and in that a linear combination with adaptive weighting factors is formed by the output signals of the recursive filter. The filter is updated by a single signal being utilized for updating the non-recursive part and the adaptive weighting factors in the recursive part. A stable filter is also obtained in this way, due to the poles of the recursive filter not being displaced.

The characterizing features of the invention are apparent from the claims.

### BRIEF DESCRIPTION OF DRAWINGS

The invention will now be described in more detail and with reference to the drawings, where

Figure 1 illustrates a known apparatus for echo cancellation,

Figure 2 illustrates an example of a desired impulse response from a filter in accordance with the invention,

Figure 3 illustrates a first embodiment example of a filter in accordance with the invention,

Figure 4 illustrates a more detailed embodiment of the filter according to Figure 3,

Figure 5 is a series of graphs giving examples of different impulse responses in certain individual filters included in the filter in accordance with the invention and

Figure 6 illustrates a second embodiment example of a filter in accordance with the invention.

### BEST MODES FOR CARRYING OUT THE INVENTION

A known apparatus for echo cancellation is illustrated in Figure 1. A digital input signal x(n) occuring on the transmission side of telecommuni-

cation equipment is applied to a two-wire to four-wire junction 2, i.e. a hybrid, which is connected to a receiver side in the telecommmunication equipment and across a two-wire line to a telephone set 4. Echo signals occur in the hybrid and in the two-wire line. The output signal to the receiver side from the hybrid 2 is denoted d(n) and consists solely of echo signals when no signal is received from the telephone set 4. This signal agrees with the above-mentioned desired signal.

The input signal x(n) is also applied to an adaptive FIR filter 1, which generates an expected echo signal y(n). An error signal e(n) is formed in summing means 3, this signal being the difference between the signals d(n) and y(n), and is utilized for updating the filter. As will be seen from the above, an FIR filter can be updated according to known methods, e.g. the LMS method. The impulse response of the filter is however generally too short for effective echo cancellation to be obtained.

In Figure 2 there is illustrated an example of a desired impulse response h(n) with relatively long extension in time, where n denotes the sequential number for the respective sample value. The impulse response can be divided into two main parts. There is first a considerable transient containing the greater part of the signal energy of the impulse. There is then a long, and substantially exponentially decaying part, a so-called tail. Negative values can also occur in the impulse response.

A first embodiment example of a filter in accordance with the invention is illustrated in Figure 3. The filter obtains a digital signal x(n) as input signal, this signal corresponding, for example, to the signal x(n) in the apparatus according to Figure 1. The input signal is applied to an FIR filter 11 directly, and to a plurality of IIR filters 13-16 after delay by a time $\tau$ in a delay means 12. The IIR filters are suitably of the first degree, and have permanently set filter coefficients having mutually differing values. The output signal from the FIR filter 11 is supplied to a summing means 22, and the output signals from the IIR filters 13-16 are each applied to their respective multiplication means 18-21. Each of the latter has an adaptive multiplication factor. These multiplication factors are assumed to have the values W0-W3, and they are adjusted in the way given below. The output signals from the FIR filter 11 and from the multiplication means 18-21 are finally added in the summing means.

In accordance with the inventive concept, the first part of the impulse response is generated in the FIR filter 11 and the second part is generated as a linear combination of the output signals from the IIR filters 13-16. The weightings in the linear combination are here determined by the adaptive multiplication factors, or weighting factors W0-W3. By suitable delay of the input signal x(n) to the IIR filters, both parts of the impulse response can be generated independently of each other. The filters in accordance with the invention thus comprise two separate filter parts, a non-recursive filter part and a recursive filter part, the output signals of which are added.

The filter output signal is denoted y(n) and is subtracted from an arbitrary desired signal d(n) in a summing means 3. A difference signal e(n) thus obtained occurs on a line 17 and is utilized both for updating the non-recursive filter part, i.e. the FIR filter 11, and the recursive filter part. Updating the latter takes place by updating the adaptive weighting factors W0-W3 of the multiplication means 18-21. The signals y(n), d(n) and e(n) and the summing means 3 agree with corresponding signals and means in Figure 1, for example, but the field of application of the filter is of course not limited to echo cancellation. For the sake of completeness, it is pointed out that updating means are required both for the FIR filter 11 and the multiplication means 18-21, these updating means being generally known in connection with digital filters.

There is shown in Figure 4 a more detailed embodiment of the filter according to Figure 3. The FIR filter 11 conventionally comprises delay means 38-40, multiplication means 34-37, and summing means 31-33. The IIR filters 13-16 are of the first degree, and each has its permanently set filter coefficient. These filters are also implemented conventionally and each comprises a summing means, e.g. 131, a delay means, e.g. 132, and a multiplication means, e.g. 133. The multiplication means are each allocated a permanently set coefficient P0-P3, which have mutually different values, and which are thus the filter coefficients of the IIR filters.

Each of the delay means 38-40 included in the FIR filter 11 delay the input signal x(n) by a sample value, and together these correspond to the delay means 12 illustrated in Figure 3. In the illustrated example, $\tau = 3$ T. Such a separate delay means is thus not required in practise but can be included in the FIR filter instead. The summing means 22 in Figure 3 is shown in Figure 4 as a plurality of separate summing means 221-224.

As will be seen from above, the difference signal e(n) is used for both updating the FIR filter 11 and the adaptive weighting factors W0-W3 of the multiplication means 18-21 in the recursive filter part. The problem of minimizing the difference signal e(n) is equal to minimizing the sum of the square of the expression $W0 \times P0^n + W1 \times P1^n + W2 \times P2^n + W3 \times P3^n - f(n)$, where n goes from zero to infinity, P0-P3 are the permanent recursive filter coefficients and f(n) is the desired impulse response. This sum has a quadratic error area with only one minimum, since the weighting factors are only present linearly in the expression. This means that the recursive filter part can be updated according to the same method as the non-recursive filter part, e.g. according to the LMS method.

Some of the advantages achieved with the filter in accordance with the invention are that the difference signal is represented by a quadratic error area, simultaneously as the difference signal represents the actual error (as opposed to an equation error structure). In addition, the recursive filter part is always stable, since the poles of the individual IIR filters are not displaced. This depends in turn on that the filter coefficients P0-P3 are permanent.

Some graphs are illustrated in Figure 5, and are examples of different impulse responses of the

individual IIR filters in the recursive part of the filter. The transfer functions of the IIR filters 13-16 are denoted in turn by h0(n)-h3(n). It is assumed that the input signal to the filters is delayed by a plurality of sample values corresponding to the length of the impulse response of the FIR filter. The filter coefficients P0-P3 are, according to the example, 0,5, 0,75, 0,875 and 0,9375. The transfer functions will then be: $h0(n) = 0{,}5^n$, $h1(n) = 0{,}75^n$ etc. Other coefficient values can of course be selected.

The part of the entire desired impulse response occurring to the left of the impulse responses illustrated in Figure 5, i.e. earlier than these, is generated in the FIR filter 11. This is adapted such that its output signal ceases when the impulse responses according to Figure 5 start. It is pointed out, however, that the number of delay means in the FIR filter included in the filter according to Figure 4 is not adapted to the graphs in Figure 5.

By linearly combining a plurality of given impulse responses in the way described above, it is possible to achieve impulse responses of very varying forms. Both positive and negative weighting factors W0-W3 can thus occur, of course. The long decaying part of the desired impulse response cannot always be imitated exactly. This does not make so much difference, however, since only a relatively small part of the energy of the entire desired impulse response is in this part. On the other hand, the first, major part of the impulse response which is generated by the FIR filter can be imitated rather precisely.

A second embodiment example of a filter in accordance with the invention is illustrated in Figure 6. Further to the means included in the filter according to Figure 3, there is also a network denoted by 50 in this filter. The network 50 includes multiplication means and summing means, which are adapted to form linear combinations of the output signals of the IIR filters 13-16. These means are connected such that the multiplication means 18 obtains the output signal from the filter 13 in an unaltered condition. The multiplication means 19 obtains the sum of the output signal from the filter 14 and the output signal from the filter 13 multiplied by a factor, and so on, according to the Figure. The linear combinations can be selected such that the input signals to the multiplications means 18, 21 will be orthogonal. These orthogonal impulse responses are then weighted by adaptive weighting factors, as with the filters according to figures 3 and 4. A change in a given weighting factor does not necessarily cause a change in the remaining weighting factors in this case. More rapid convergence is thus obtained. The number of calculations increases somewhat, however.

The filter in accordance with the invention can be used in different connections, when a relatively long impulse response is desired and not only for adaptive echo cancellation. Of course, the number of IIR filters may be both more or less than just four, as illustrated in the examples. The implementation of the FIR and IIR filters can also be different from what has been shown in the examples. Neither is it necessary to delay the input signal to the IIR filters. However, the delay results in that the first part of the desired impulse response is generated solely by the FIR filter, and that the second part of the response is generated solely by the recursive filter part.

## Claims

1. Adaptive, digital filter including a non-recursive part and a recursive part, characterized in that the recursive part includes a plurality of branches each having its separate, permanently set recursive filter (13-16) which have mutually different impulse responses, and individual multiplication means (18-21), with an adaptive multiplication factor (W0-W3), in that the recursive part also includes summing means (22, 222-224) which, together with said multiplication means (18-21), are adapted to form a linear combination of the output signals of the recursive filters (13-16), and in that the filter is adapted for updating by a single signal (e(n)) being used for updating the non-recursive part (11) and the adaptive multiplication factors (W0-W3) of said multiplication means (18-21) in the recursive part.

2. Adaptive filter as claimed in claim 1, characterized in that the recursive filters (13-16) are of the first degree.

3. Adaptive filter as claimed in claim 2, characterized in that the filter also includes summing means (22, 221-224) for summing the output signal of the nonrecursive part (11) and said linear combination.

4. Adaptive filter as claimed in claim 3, characterized in that the filter also includes a delay means (12, 38-40) adapted such that an input signal (x(n)) applied to the filter is applied to the recursive filters (13-16) after a predetermined delay.

5. Adaptive filter as claimed in any one of claims 1-4, characterized in that the filter also includes a network (50) inserted between the recursive filters (13-16) and said multiplication means (18-21), and which is adapted to form linear combinations of the output signals of the recursive filters (13-16).

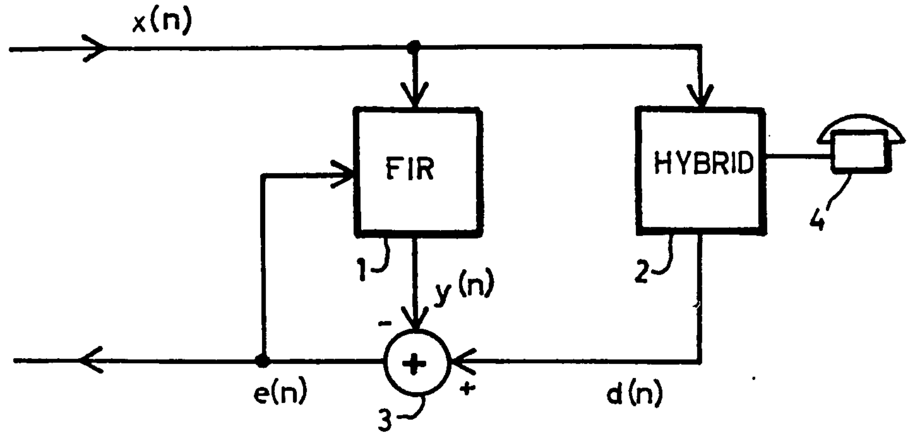



Fig.1

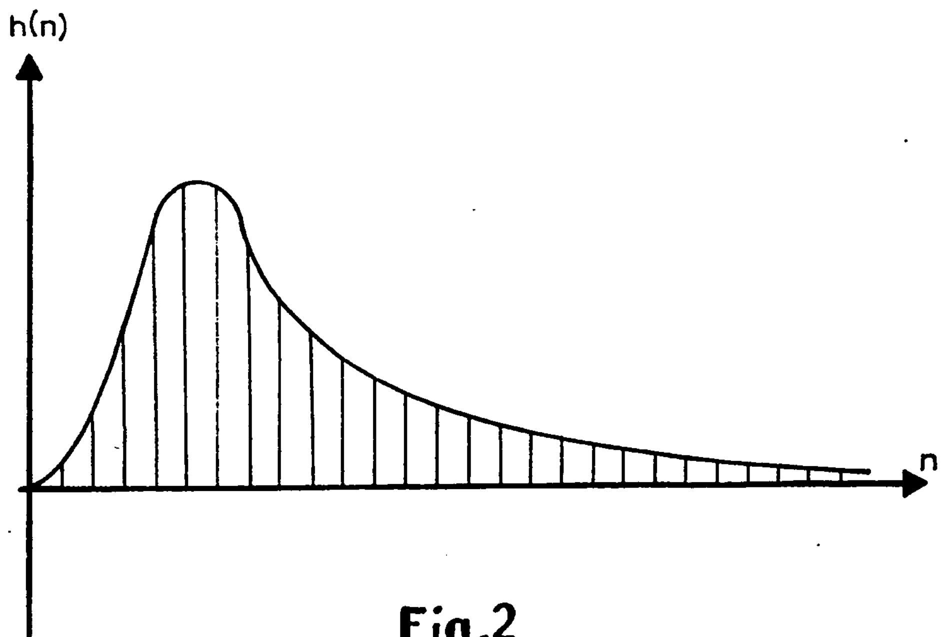



Fig.2

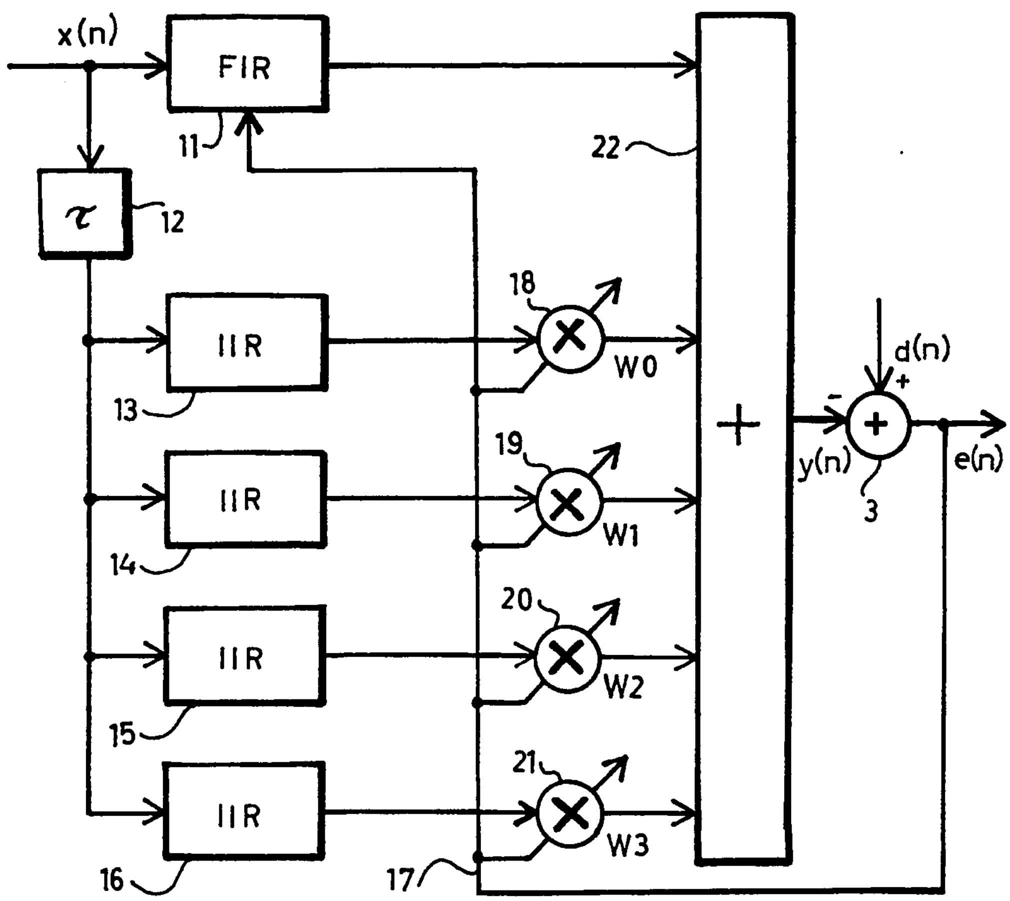
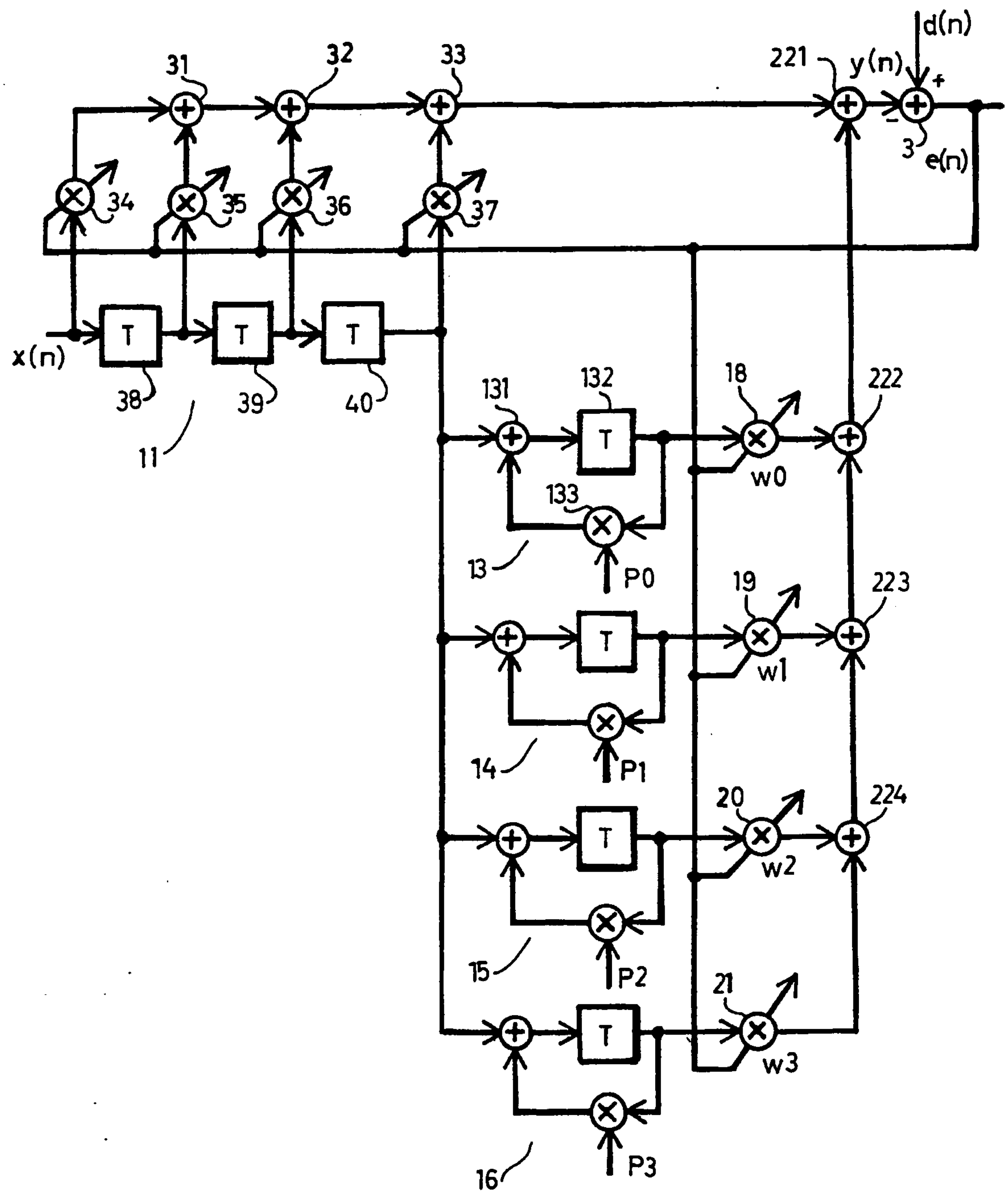

x(n)
FIR
11
τ
12
22
18
19
20
21
IIR
IIR
IIR
IIR
13
14
15
16
17
W0
W1
W2
W3
+
d(n)
+
-
y(n)
3
e(n)

Fig.3


d(n)
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32
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3
e(n)
34
35
36
37
x(n)
T
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T
38
39
40
11
131
132
18
222
133
w0
13
P0
19
223
14
w1
P1
20
224
w2
15
P2
21
w3
16
P3

Fig 4

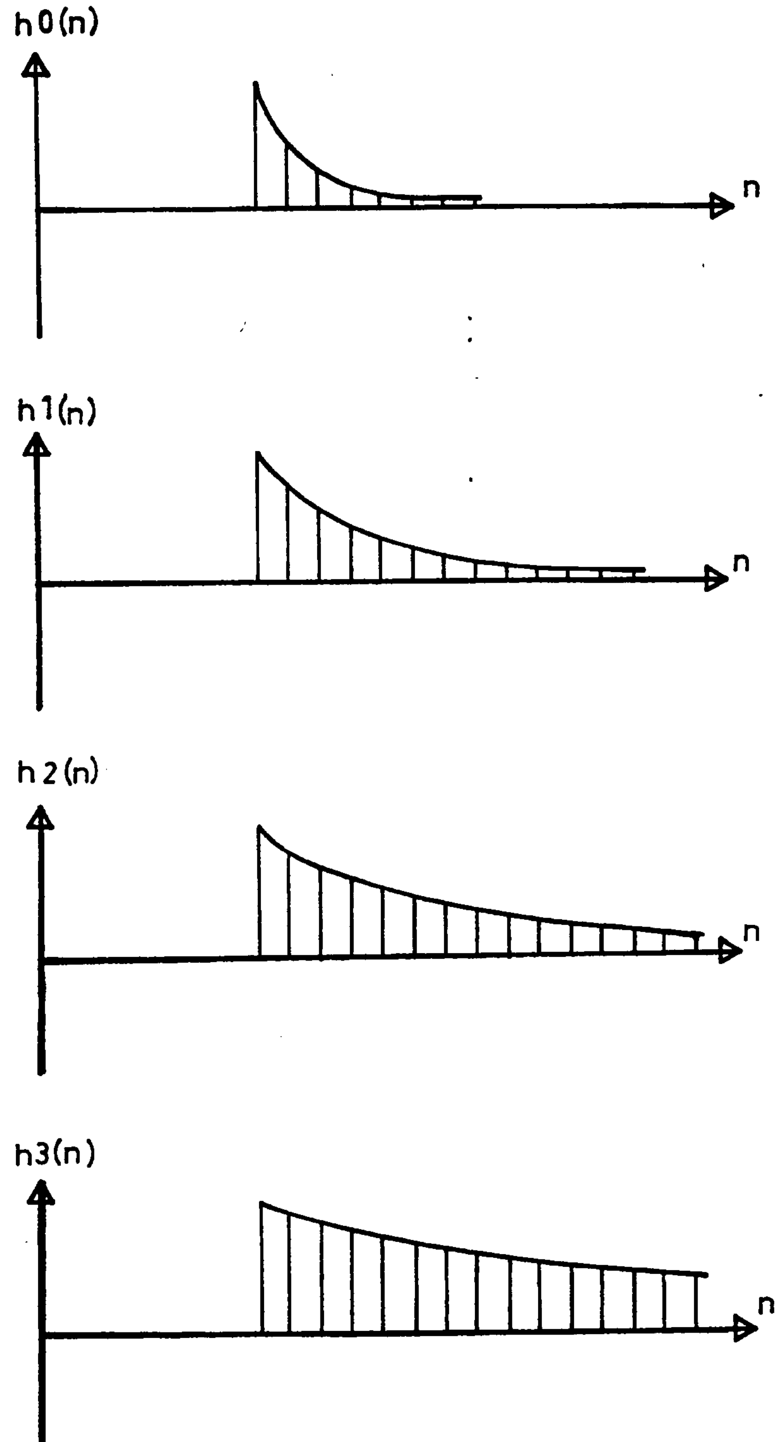


Fig.5

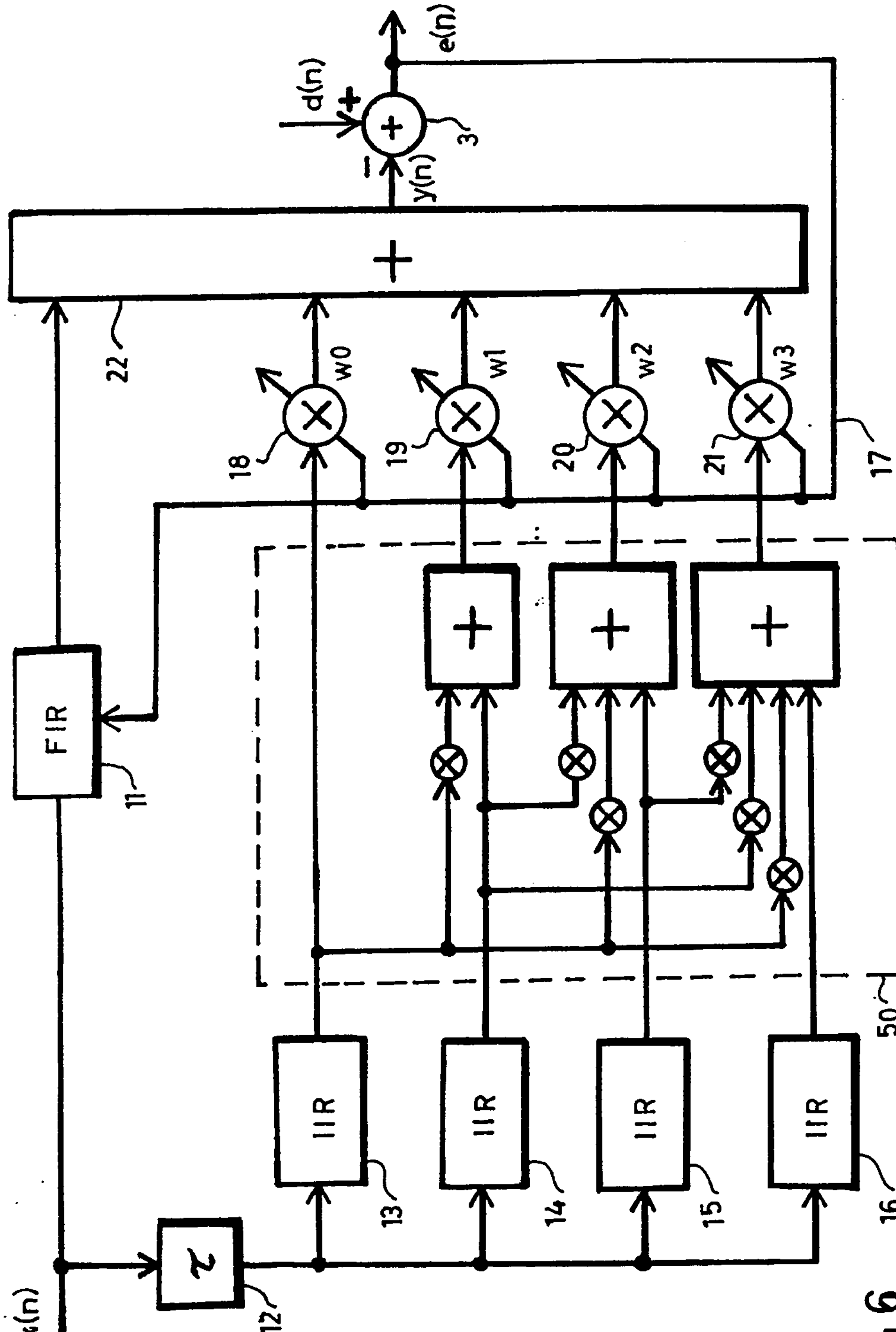
e(n)
d(n)
+
+
−
y(n)
3
+
22
w0
w1
w2
w3
18
19
20
21
17
FIR
11
+
+
+
50
IIR
IIR
IIR
IIR
13
14
15
16
x(n)
12

Fig.6

European Patent Office

# EUROPEAN SEARCH REPORT

Application number

EP 89850107.7

## DOCUMENTS CONSIDERED TO BE RELEVANT

| Category | Citation of document with indication, where appropriate, of relevant passages | Relevant to claim | CLASSIFICATION OF THE APPLICATION (Int Cl 4) |
|---|---|---|---|
| A | GB-A- 2 102 255 (INTERNATIONAL STANDARD ELECTRIC CORPORATION) *Abstract* | 1-5 | H 03 H 21/00<br>H 04 B 3/23 |
| A | US-A- 4 520 491 (RAULIN ET AL) *Abstract* | 1-5 | |
| A | US-A- 4 539 690 (SPEIDEL) *Whole document* | 1-5 | |
| A | EP-A2-0 253 583 (OKI ELECTRIC INDUSTRY COMPANY LIMITED) *Whole document* | 1-5 | |
| A | DE-A1-3 610 382 (ANT NACHRICHTENTECHNIK GMBH) *Whole document* | 1-5 | |
| | | | **TECHNICAL FIELDS SEARCHED (Int. Cl.4)** |
| | | | H 03 H<br>H 04 B |

The present search report has been drawn up for all claims

| Place of search | Date of completion of the search | Examiner |
|---|---|---|
| STOCKHOLM | 14-08-1989 | BENGTSSON R. |

CATEGORY OF CITED DOCUMENTS

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